

UDC 621.397.743
621.395.665.1



RESEARCH DEPARTMENT



REPORT

**A p.c.m. sound-in-syncs
distribution system:
syllabic companding**

No. 1969/48

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Work covered by this report was undertaken by the BBC Research Department for the BBC and the ITA

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A P.C.M. SOUND-IN-SYNCS DISTRIBUTION SYSTEM: SYLLABIC COMPANDING

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A P.C.M. SOUND-IN-SYNCS DISTRIBUTION SYSTEM: SYLLABIC COMPANDING

SUMMARY

The p.c.m. 'sound-in-syncs' system described in an earlier report¹ incorporates a frequency-dependent syllabic compander, in which an out-of-band pilot tone is used to control the expander gain. The present report describes the essential features of this compander, gives performance data obtained with experimental equipment and suggests some possible future improvements.

1. INTRODUCTION

In an earlier report¹ an account was given of a 'sound-in-syncs' system, in which the sound associated with a television programme is distributed to transmitters in the form of p.c.m. signals incorporated in the video waveform.

In order to achieve maximum immunity of the p.c.m. signal against noise and distortion in the vision links and to avoid interference between sound and picture, the number of bits per word in the p.c.m. sound coding is restricted to ten; the level of quantizing noise in a ten-bit code is above the acceptable limit for broadcast-quality sound, but in the sound-in-syncs system an adequate overall signal-to-noise ratio is achieved by the use of a syllabic compander.

The mode of operation of this compander and the considerations which determine the optimum circuit parameters have been dealt with in some detail in an earlier report.² The present report briefly recapitulates the principles involved and describes the experimental compander constructed to meet the requirements of the sound-in-syncs system; performance data are given for various parts of the equipment and some possibilities for future improvement are indicated.

2. DESCRIPTION OF EQUIPMENT

2.1. Principle of Operation

In the compander to be described, a quick-acting audio-frequency limiter is introduced at the sending terminal ahead of the analogue-to-digital converter (a.d.c.) of the p.c.m. system in order to prevent over-modulation; this limiter is arranged to reduce gain

whenever the level of the incoming signal exceeds the maximum for which the system is designed. High-frequency pre-emphasis is applied ahead of the limiter, and the maximum level of the incoming signal is made just sufficient to reach the limiting point when the proportion of high-frequency components is too small for the pre-emphasis to have any effect. Whenever the signal contains predominant high-frequency components, the maximum level is raised by the pre-emphasis to above this point; the limiter then operates and the gain of the system is temporarily reduced.

In addition to the pre-emphasized signal applied to the limiter input, an out-of-band pilot tone is injected at a low and constant level, and both are subject to the same variations in gain. At the receiving terminal of the p.c.m. system, the reconstructed audio-frequency signal, which now includes the pilot tone, is applied to a second limiter. This limiter is designed to regulate its gain in such a way that the level of pilot tone at the output is held constant. Changes in the gain of the first limiter are thus compensated by equal and opposite changes in the gain of the second; the two limiters function as compressor and expander respectively and the overall gain of the system is held constant. Finally, the expanded signal is de-emphasized, the noise level being reduced in the process.

The use of pre- and de-emphasis in the manner described prevents the quantizing noise from the p.c.m. system from being heard when the expander gain increases on arrival of a signal; the pre-emphasis characteristic is such that the compander operates only when the signal contains a sufficient proportion of high-frequency components to mask the rise in noise, and the so-called 'hush-hush' effect — a common defect of companders — is thereby avoided.

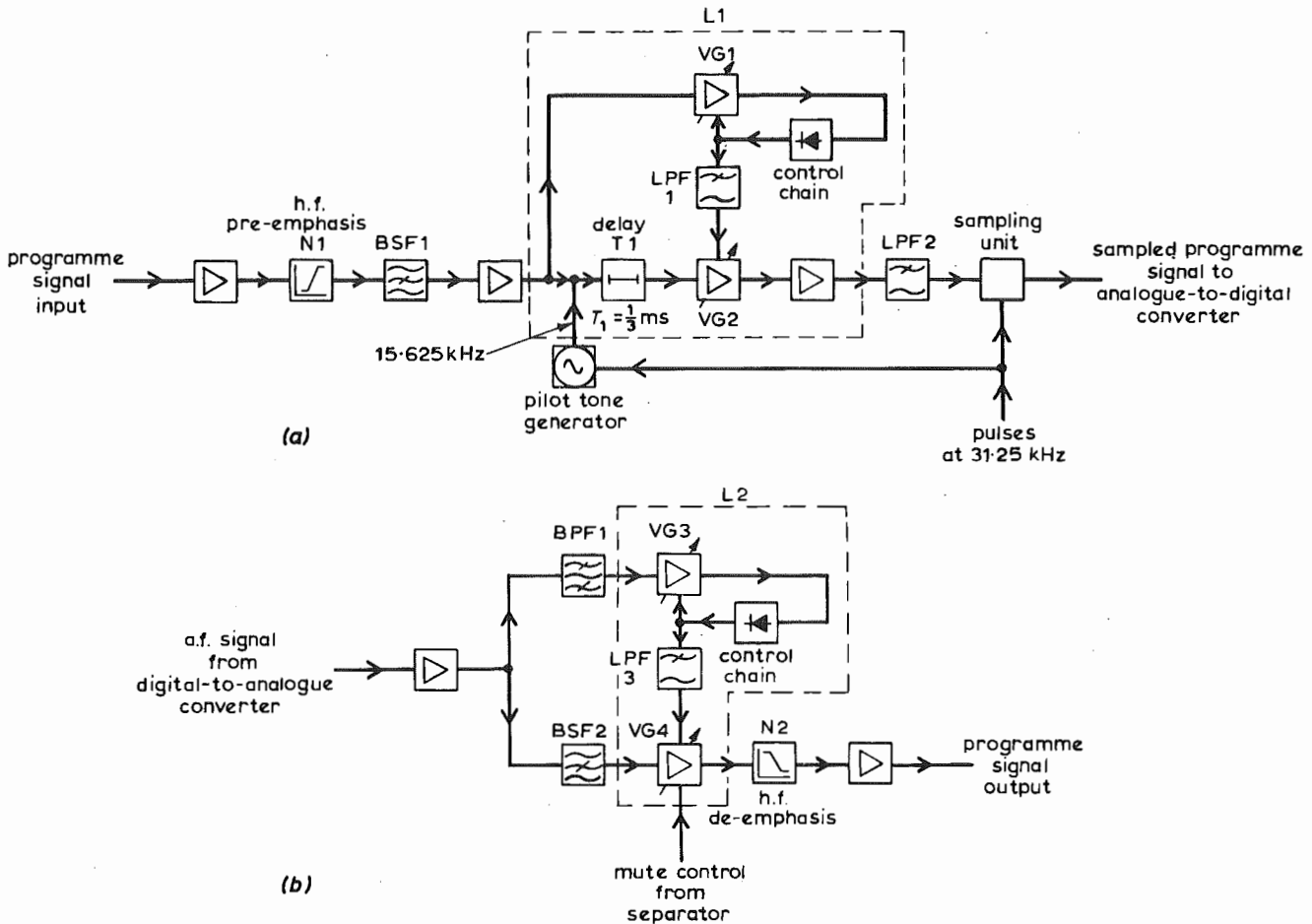


Fig. 1 - Schematic diagram of compandor used with experimental sound-in-syncs system

(a) Sending Terminal

(b) Receiving Terminal

2.2. Sending Terminal

2.2.1. General

Fig. 1(a) shows in block schematic form the sending terminal equipment of the experimental compandor. At the sending terminal, incoming audio-frequency signals are pre-emphasized in network N1, which has the amplitude/frequency characteristic shown in Fig. 2 curve (a), and are applied to limiter L1 through band-stop filter BSF1, which rejects all components within a narrow band centred on the pilot-tone frequency of 15.625 kHz. The pilot tone is derived from a train of pulses at 31.25 kHz – the sampling frequency of the a.d.c.

For reasons explained in an earlier report,³ limiter L1 is provided with two identical variable-gain stages VG1 and VG2. A control voltage, derived by rectification of the output signal from VG1, is made to vary the gain of that stage in such a way as to prevent the output signal from exceeding a prescribed amplitude; the operating time of the control loop is short enough to produce equilibrium within a few tens

of microseconds after the application of a signal.

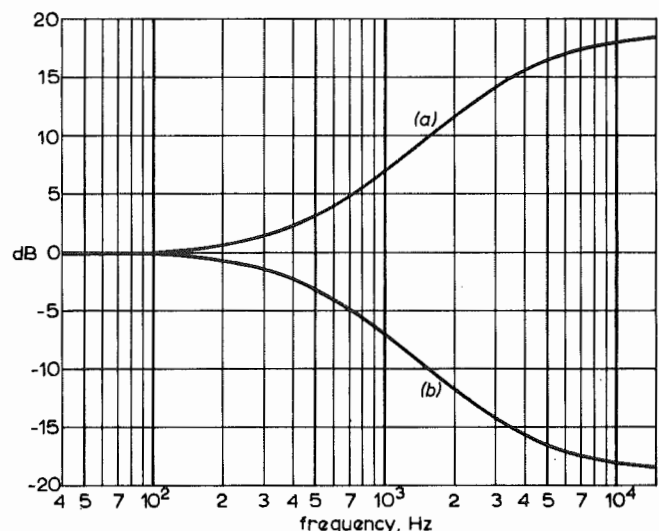


Fig. 2 - Pre- and de-emphasis characteristics

(a) Pre-emphasis characteristic
(b) De-emphasis characteristic

The control voltage applied to VG1 is also applied to VG2 via a low-pass filter LPF1, having a nominal cut-off frequency of 1.5 kHz; the function of LPF1 is to restrict the rate of change of gain of VG2 which, if excessive, could lead to the production of audible distortion products. After the arrival of a signal at VG1, a period of about 1/3 ms elapses before the gain of VG2 is reduced to the appropriate value; to avoid transient overloading of the a.d.c. during this period, the signal reaching VG2 is delayed by the same amount in network T1.

2.2.2. Band-stop Filter BSF 1

BSF1 is a five-branch Chebyshev filter which attenuates by about 75 dB any programme signal components appearing in the range $15.625 \text{ kHz} \pm 500 \text{ Hz}$ while producing negligible attenuation at 14 kHz, the specified upper-frequency limit of the programme signal in the p.c.m. system.

2.2.3. Pilot Tone Generator

The pilot tone generator consists of a flip-flop actuated by pulses at 31.25 kHz, the sampling frequency of the p.c.m. system. The output from this stage, a constant-amplitude square wave at 15.625 kHz, is filtered to remove harmonics, and the phase of the resulting tone is so adjusted that sampling in the a.d.c. takes place at the point of maximum amplitude in each half-cycle.

No difficulty was experienced in keeping the day-to-day stability of the pilot tone level within $\pm 0.05 \text{ dB}$; no adjustment of level or phase was found necessary during the field trials of the sound-in-sync equipment.

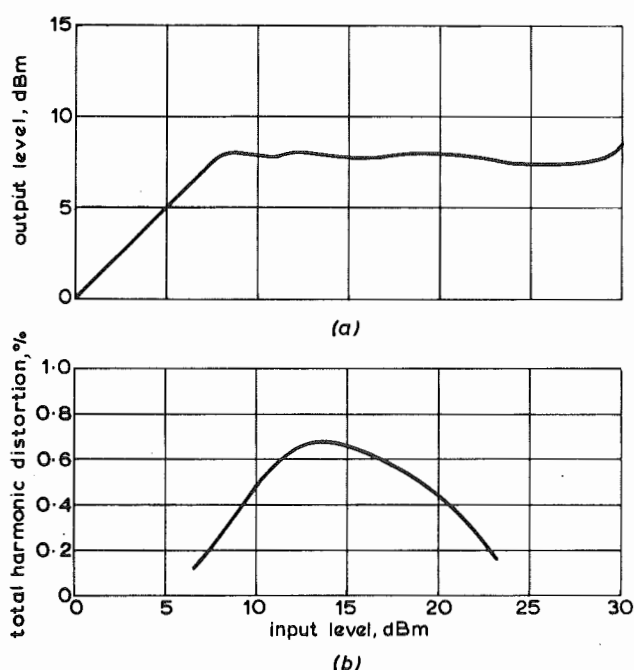


Fig. 3 - Limiter L1

(a) Output/-input characteristic (b) Total harmonic distortion

2.2.4. Limiter L1

Fig. 3 curve (a) shows the output/input characteristic of limiter L1. Departures from the ideal flat-topped curve are due to mismatch between the nominally identical variable-gain stages VG1 and VG2. Gain variation is effected by the use of F.E.T.s as variable shunt resistance elements; by selection of these F.E.T.s, the output/input characteristic beyond the limiting point is kept flat within $\pm 0.3 \text{ dB}$ over the working range of about 20 dB.

Fig. 3 curve (b) shows the total harmonic distortion of the limiter at 1 kHz as a function of input signal level. The distortion is almost entirely due to non-linearity of the F.E.T. in VG2 and is greatest when the impedance of this F.E.T. is about the same as that of the circuit from which it is fed, i.e. when the gain of VG2 is about 6 dB below its maximum value.

2.3. Receiving Terminal

2.3.1. General

At the receiving terminal, shown in Fig. 1(b) the audio-frequency output from the digital-to-analogue converter (d.a.c.) is applied to band-pass filter BPF1, which passes only a narrow band of frequencies centred on the pilot tone, and to band-stop filter BSF2, which rejects the band of frequencies passed by BPF1 but passes the programme signal.

The limiter L2 which, as previously stated, performs the function of an expander, is similar in construction to L1, having a pair of identical variable-gain stages VG3 and VG4 corresponding in function to VG1 and VG2 respectively, but no signal delay network is provided. The pilot tone from BPF1, the level of which varies according to the gain of L1, is applied to VG3. Matters are so arranged that the lowest level of pilot tone incoming is sufficient to operate the control chain of L2; for higher levels the output from VG3 remains constant. The programme signal from BSF2 is applied to VG4, the gain fluctuations of which offset those produced in VG2, and thence via a de-emphasis network N2, having the amplitude/frequency characteristic shown in Fig. 2 curve (b), to the output. VG4 also contains an overriding gain control circuit, actuated by a signal from the video separator circuit⁴ in the receiving terminal of the sound-in-sync system, by which the outgoing signal can be momentarily muted to minimise the audible effect of digit errors in the p.c.m. system; such errors may occur sporadically as a result of irregularity in the line sync pulses or through interference on the vision link.

2.3.2. Band-Pass Filter BPF1

BPF1 is a five-branch Chebyshev filter having a pass-band of $15.625 \text{ kHz} \pm 500 \text{ Hz}$ and giving an

attenuation of 75 dB at 14 kHz. The combined effect of this filter and of the band-stop filter BSF1 at the sending terminal is to ensure that the level of any programme signal components reaching the input of VG3 is at least 30 dB below (3%) that of the pilot tone at that point and thus cannot affect the outgoing signal amplitude by more than $\pm 1/3$ dB.

The dynamic characteristics of the compandor system depend among other things on the time response of BPF1 to changes in signal level. Fig. 4 curve (a) shows the envelope of a pilot tone being reduced in amplitude by 18 dB over a period of $1/3$ ms — the maximum rate-of-change of gain that the sending terminal could produce. Fig. 4 curve (b) shows the voltage at the output of BPF1 when the voltage in curve (a) is applied to the input. It will be seen that the filter introduces a delay of about 1 ms, restricts the rate-of-change of amplitude and causes slight beating of the signal.

2.3.3. Band-Stop Filter BSF2

The stop-band attenuation required for BSF 2 depends on the maximum level at which the pilot tone may be allowed to appear at the output of the system without danger of its being audible. Time did not permit a reliable figure to be arrived at by subjective tests, but it was estimated from available data^{5,6} that a filter identical with BSF1, having more than 70 dB attenuation in the stop band, should give ample suppression, and this proved to be the case.

2.3.4. Limiter L2

The circuit of L2, although similar in form to that of L1, was adopted for different reasons; in this case the main purpose in using duplicate variable-gain stages is to avoid having BPF1 within an a.g.c. loop² where the delay introduced by that filter would make it difficult to achieve adequate speed of operation while avoiding instability.

Because the delay network incorporated in L1 is omitted in L2, a signal can appear at VG4 before the gain of this stage has assumed the appropriate value. The amount of this timing discrepancy depends on the delay in LPF3 (amounting, as in the corresponding filter LPF1 at the sending terminal, to $1/3$ ms), the delay and the restricted rate-of-change imposed by BPF1 (already illustrated in Section 2.3.2), together with the operating time of the a.g.c. loop formed by VG3 plus its control chain. The speed of operation of the a.g.c. is limited by the control rectifier smoothing time constant which, to avoid the possibility of ripple components intermodulating with the signal in VG4, cannot be made less than about 0.7 ms. The total time lag from all these causes amounts to a few milliseconds, but the effect of the expander action being delayed by this amount has been found in practice to be inaudible.²

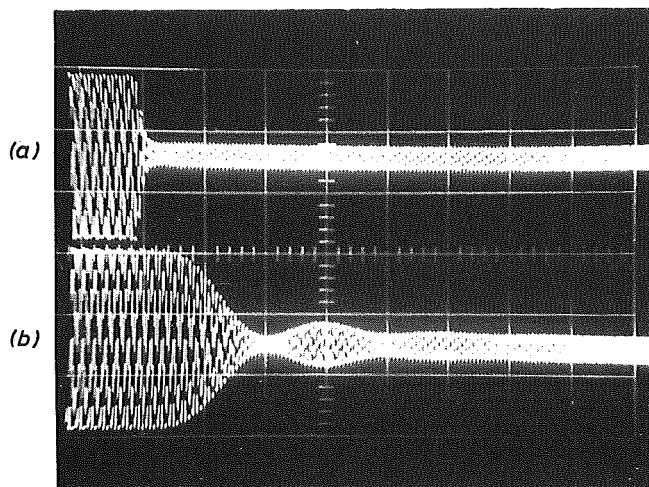


Fig. 4 - Amplitude response of filter BPF1 to rapid change in level of pilot tone

(a) Input (b) Output
Time scale 1 ms/cm

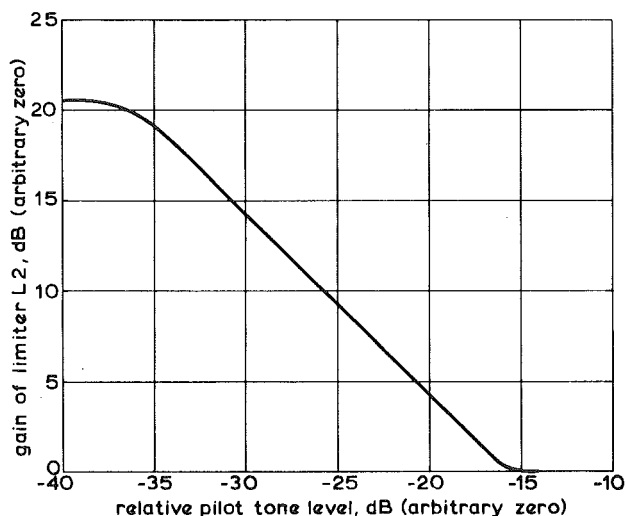


Fig. 5 - Limiter L2

Variation of gain with level of pilot tone

Fig. 5 shows the gain of L2 as a function of incoming pilot-tone level. Ideally the gain should fall by 1 dB for every 1 dB increase in pilot tone level; as with limiter L1, errors in gain can result from mismatch between the two variable-gain stages, but by selection of F.E.T.s can be kept within $\pm 1/3$ dB over a 20 dB range.

In L2, as in L1, the non-linear distortion is greatest when the gain is about 6 dB less than its maximum value, but in this condition, the signal level is lower than in L1; for the same degree of distortion the variable-gain element can therefore be operated at a higher maximum signal level and the effect of circuit noise thereby reduced. Fortunately, since the gain of L1 and L2 vary with signal level in opposite directions over a range of 20 dB, the conditions for greatest distortion never occur in both at the same time.

3. OVERALL PERFORMANCE

3.1. General

The data given in the following sections relates, where not otherwise stated, to the sending and receiving terminals of an experimental compandor operated back-to-back without the associated p.c.m. system.

3.2. Output/Input Characteristic (Tracking Accuracy)

The output/input characteristic measured at 1 kHz shows less than $\pm 1/3$ dB gain variation with level. This figure is determined by the accuracy of matching of the two variable-gain stages in limiter L2, but is independent of the corresponding matching in limiter L1.

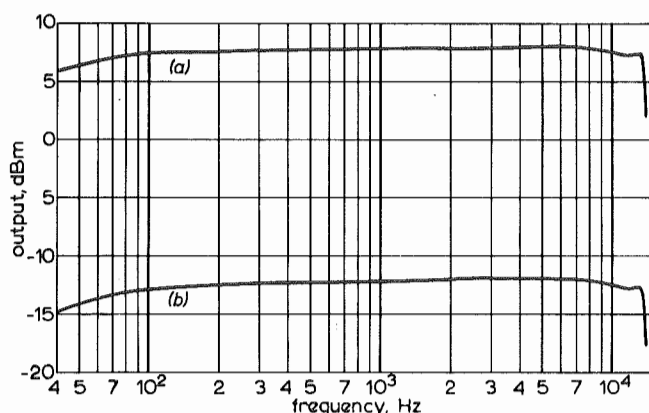


Fig. 6 - Amplitude/frequency characteristic of compandor (typical)

- (a) With input signal 20 dB below maximum level.
(b) With input signal at maximum level

3.3. Amplitude/Frequency Characteristics

Fig. 6 shows the variation in output level from the compandor system with frequency when the input level was kept constant.

Curve (a) was obtained with an input signal at a level 20 dB below the rated maximum for the system — too low to operate the compandor.

Curve (b) was obtained with an input signal at the rated maximum level so that the compandor was operative. The slight difference in shape of curves (a) and (b) is due to the mismatch of some $\pm 1/3$ dB between the nominally identical variable-gain elements in VG3 and VG4, already referred to in Section 3.2; when the compandor is operating, this mismatch causes the overall gain of the system to vary a little with the signal level applied to L1, which in turn varies with frequency in the course of the measurement because of the pre-emphasis at the sending terminal. This effect does not alter the relative gain of the system

for signal components at different frequencies simultaneously applied, i.e., the difference between curves (a) and (b) does not signify any change in the spectrum of the output signal with level.

3.4. Signal-to-Noise Ratio

For the compandor alone, the ratio of peak signal to peak weighted* noise is 74 dB.

When the compandor is used in conjunction with the p.c.m. system, the improvement in signal-to-quantizing noise ratio which can be achieved is equal to the noise reduction produced by the de-emphasis network N2; this figure has been estimated by subjective tests to be 15 dB. The ratio of peak signal to peak weighted noise for a 10-bit p.c.m. system is 53 dB⁷, so that in theory an overall signal-to-noise ratio of 68 dB should be possible. This figure is diminished in practice by 1 dB through the self-generated noise of the compandor; referred to above, and by a further 1 dB representing a 10% reduction in input signal level to accommodate the pilot tone; the resulting signal-to-noise ratio is thus 66 dB and the net improvement effected by the compandor, 13 dB.

3.5. Non-linear Distortions

Fig. 7 shows the total harmonic distortion of this compandor system at 1 kHz as a function of signal level. It should be noted that the values given include the effects of the de-emphasis at the receiving terminal.

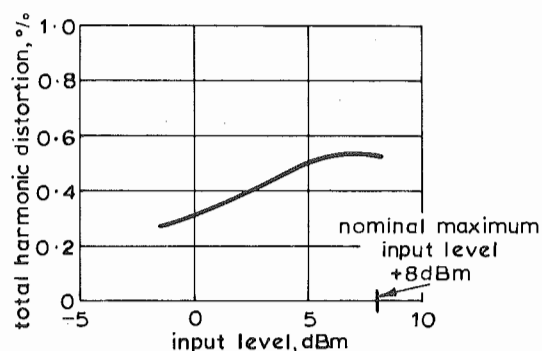


Fig. 7 - Total harmonic distortion of experimental compandor at 1 kHz as function of input level

3.6. Dynamic Characteristics

To illustrate the action of the compandor under transient conditions Fig. 8 shows the output from the system following the sudden application of a 5 kHz signal to the input. The initial irregularities on the signal envelope are attributable to ringing of the band pass filter BPF1 and to dispersion in the delay network. As indicated in Section 2.3.4 the delay in reach-

* 1949 CCIF weighting.

ing the steady signal amplitude is inappreciable in practice.

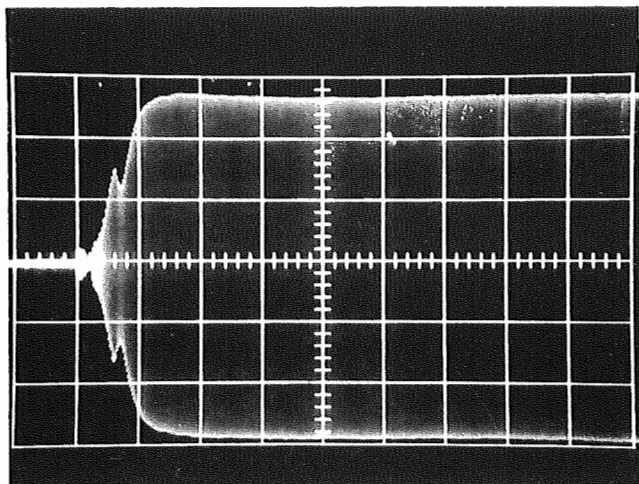


Fig. 8 - Overall amplitude response of compandor to suddenly applied 5 kHz tone

Time Scale 5 ms/cm

4. FUTURE DEVELOPMENT

As a result of experience with the experimental compandor some possible improvements suggest themselves for the future.

The operating range of L1 and L2 is restricted to 20 dB, which is little more than the amount of the pre-emphasis at the sending terminal. Thus, an incoming signal having predominant high-frequency components could utilise nearly the whole range of the compandor, leaving little to spare for the case where the prescribed maximum input is accidentally exceeded. The working range of both L1 and L2 could be extended by increasing the impedance of the circuit across which the FET used as a variable-resistance element is shunted, but the thermal noise level in the system would then be increased. It would not be possible with the present circuit to restore the situation by a corresponding increase in signal level without producing an unacceptable degree of waveform distortion through the non-linear characteristics of the F.E.T. In a final design, however, consideration could be given to the artifice^{3,8} of reducing distortion by adding a proportion of the output signal from each F.E.T. to the corresponding control voltage.

Matching of the F.E.T.s in L1 and L2 could be made easier by making the individual control voltages

as well as the working points, adjustable; this artifice is employed in the limiter type AM6/7 and could be incorporated in future compandor equipment.

5. CONCLUSION

The experimental compandor described has functioned satisfactorily as part of the sound-in-synchs system and developed no faults during the field trial of the latter. It is thought that fully-engineered equipment using the same principles should be capable of meeting the requirements of the television service.

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